IN THE SPECIFICATION

On page 6, line 16 to page 7, line 24 replace the entire paragraph with the paragraph below, which show the changes made.

A method is described below for calculating a human voiced speech excitation function. The movement (position versus time) of the vocal tract ~~a tracheal wall~~ is determined using an electromagnetic sensor or equivalent, and the position is translated to pressure by determining the times of largest change in the movement waveform using a derivative, or differential, of the movement waveform. Pulses of various amplitude and width are placed at these times, and the result is shown to contain the same frequency information as the movement signal, although it can be described with considerably fewer parameters. The excitation function so produced is shown to lead to a better model of the vocal tract than is typically available using standard acoustic-only processing. The excitation function is also useful for calculating a variety of speech parameters with great accuracy, some of which are not available with conventional technology.

On page 8, lines 8 through 16, replace the entire paragraph with the paragraph below, which show the changes made.

Figure 2 is a block diagram of a speech signal processing system 200, under one alternate embodiment. The system 200 includes a glottal-area electromagnetic micropower sensor (GEMS) 20 and microphones 10, including microphone 1 and microphone 2. The GEMS sensor 20 provides signals or information used by the system to generate information including pitch, processing frames, and glottal cycle information 204, and excitation functions 206. The microphones 10 provide signals that the system uses to produce cleaned ~~clean~~ audio 70 and voicing/unvoicing information 210. Transfer functions 208 are produced using information from both the GEMS sensor 20 and the clean audio 70.

On page 20, lines 9 through 16, replace the entire paragraph with the paragraph below, which show the changes made.

In the standard model of speech production, the throat, mouth, nose, and other articulators act as a filter to shape the excitation function into the desired sound. The transfer function represents that filter. If the excitation function is filtered by the transfer function, the resulting signal (if the excitation and transfer function are good approximations) will be very close to the original speech. Typical prior art speech systems usually determine their transfer functions based on linear ~~liner~~ predictive coding (LPC) algorithms, which use no excitation function signal at all and cannot fully model the speech.

On page 20, line 17 to page 21, line 2, replace the entire paragraph with the paragraph below, which show the changes made.

In determining the transfer function, the excitation function and output are calculated or recorded using the methods described above. Then, standard signal processing system identification techniques know in the art ~~are~~ may be applied to the results to determine the transfer function. Mathematically, in the z domain

$$TF(z) = \frac{O(z)}{EF(z)},$$

where O(z) is the z-transform of the output and EF(z) is the z-transform of the excitation function. The signal processing system identification techniques used include least-mean squared (LMS) adaptive algorithms, power spectral division, and many others.